Sipura Technology, Inc.

SPA-2000 User Guide

December 2003

Sipura SPA-2000 User Guide v1.0.3.doc

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Sipura Technology, Inc.
SPA-2000 User Guide

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1. Product Description

This guide describes basic administration and use of the Sipura Technology SPA-2000 phone adapter – an intelligent low-density Voice over IP (VoIP) gateway. The SPA-2000 enables carrier class residential and business IP Telephony services delivered over broadband or high-speed Internet connections. By intelligent we mean the SPA-2000 maintains the states of all the calls it terminates. It is capable of making proper decisions in reaction to user input events (such as on/off hook or hook flash or enhanced services codes, i.e. *69) with little or no involvement by a middle-man server or media gateway controller.

1.1. SPA-2000 Hardware Overview

The SPA-2000 has one of the smallest form factors on the market. It can be installed in minutes as a table-top or wall mount CPE device. Figures Figure 1, Figure 2, Figure 3 and Figure 5 show the front, rear, left side and right side of the SPA-2000, respectively.

Figure 1 – SPA-2000 Front

Figure 2 – SPA-2000 Rear

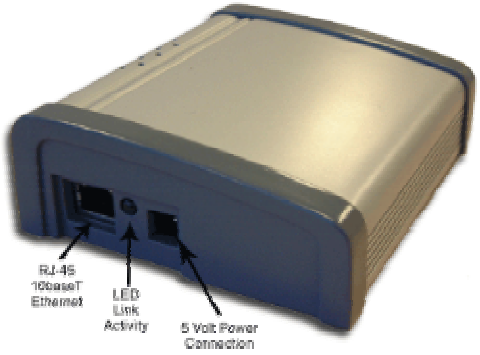

Figure 3 – SPA-2000 Left Side

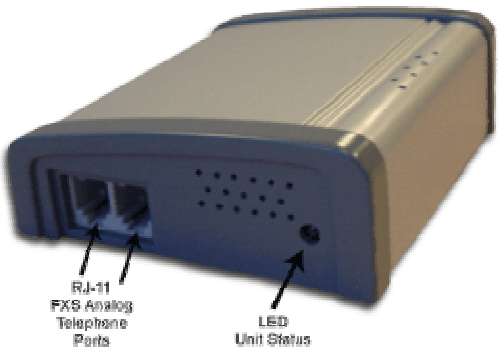

Figure 4 – SPA-2000 Right Side

The SPA-2000 has the following interfaces for networking, power and visual status indication:

1. Two (2) RJ-11 Type Analog Telephone Jack Interfaces (Figure 4, above):

These interfaces accept standard RJ-11 telephone connectors. An Analog touchtone telephone or fax machine may be connected to either interface. If the service supports only one incoming line, the analog telephone or fax machine should be connected to port one (1) of the SPA-2000. Port one (1) is the outermost telephone port on the SPA-2000 and is labeled “Phone 1.”

2. One LED for Unit Status (Figure 4, above):

This LED indicates status via the following behaviors:

ON – LED remains solid on

OFF – LED remains solid off

LONG (Long On) – 3.0s on, 1s off continuously

FAST – 0.125s on, 0.125s off continuously

SLOW – 0.5s on, 0.5s off continuously

VSLO (Very Slow) – 1.0s on, 1.0s off continuously

HB (Heart Beat) – 0.125s on, 0.125s off, 0.125s on, 1s off continuously

3. One Ethernet 10baseT RJ-45 Jack Interface (

Figure , above):

This interface accepts a standard or crossover Ethernet cable with standard RJ-45 connector. For optimum performance, Sipura Technology recommends that a Category 5 cable or greater be used in conjunction with the SPA-2000.

4. One LED for Data Link and Activity (

Figure , above):

This LED indicates status via the following behaviors:

ON – LED remains solid on

OFF – LED remains solid off

FAST – 0.125s on, 0.125s off continuously

SLOW – 0.5s on, 0.5s off continuously

Variable Blink – LED blinks according to packet traffic activity

5. One 5 Volt Power Adapter Interface (

Figure , above)

This interface accepts the SPA-2000 power adapter that came with the unit. Sipura Technology does not support the use of any other power adapters other then the power adapter that was shipped with the SPA-2000 unit.

1.1.1. Status LED Description:

The SPA-2000 Status LED is used to indicate the current operation status of the SPA unit. The state is represented by a special blinking pattern of the Status LED (next to the RJ-11 ports on the right side of the device). The below table describes the various modes and behaviors of the SPA-2000 in relation to the Status LED and the handset behavior.

Status Description	**LED Blink Rate**	**Default Handset Behavior**
Normal Operation – Both Lines on hook	OFF	Normal
Normal Operation – Either line off hook	ON	Normal
Downloading new firmware	SLOW	Silent
Writing firmware to flash	FAST	Silent
Looking for DHCP Server	SLOW	Silent
No DHCP Server	SLOW	Silent
IP Address Conflict	SLOW	Silent

Unknown DHCP Error	SLOW	Silent

Note: The Link LED will blink on transmit and receive (TX/RX) of packets. The LED will display solid off if no link is available. The LED will display solid on if link is up but no TX/RX activity is present.

Important Warning: Do not disrupt the power to the SPA-2000 while the Status LED is blinking FAST.

2. Installation Overview

Please check to make sure that you have the following package contents:
1. Sipura Phone Adapter Unit
2. Ethernet Cable
3. SPA-2000 QuickGuide
4. 5 Volt Power Adapter

You will also need:
1. One or Two Analog Touch Tone Telephones (or Fax Machine)
2. Access to an IP Network via an Ethernet Connection

Please observe the following steps to install the SPA-2000.

From the Left Side of the SPA-2000:
1. Insert a standard RJ-45 Ethernet cable (included) into the LAN port.
2. Insert the power adapter cable into the 5V power adapter cable receptacle.
   Ensure that the power adapter jack is snugly attached to the SPA.

From the Right Side of the SPA-2000:
1. Insert a standard RJ-11 telephone cable into the Phone 1 port.
2. Connect the other end of the cable to an analog telephone or fax machine.
3. Insert a standard RJ-11 telephone cable into the Phone 2 port (Optional).
4. Connect the other end of the cable to an analog telephone or fax machine.
   Note: Do not connect RJ-11 telephone cable from the SPA to the wall jack
   to prevent any chance of connection to the circuit switched telco network.

You may now insert the plug end of the power adapter into a live power outlet which will power up the SPA-2000.

3. Software Configuration

3.1.1.1. Firmware Upgrade

The SPA-2000 is firmware upgradeable via TFTP or via an executable PC program obtained from Sipura Technology or an authorized distributor/reseller of Sipura Technology products.

Please contact the company from whom you purchased your SPA-2000 for access to Sipura Technology firmware upgrades.

Firmware Upgrade via TFTP:
Firmware designed for TFTP upgrades are released as single binary files, which contain all the modules pertaining to any one release version. By convention, the firmware loads are named with the extension ".bin" (e.g. spa.bin)

The SPA-2000 can be configured to upgrade to a specific version, possibly staging through intermediate releases, if necessary. This process can be automated for a pool of devices through configuration profile parameters.

Alternatively, an individual SPA-2000 can be directed to perform an upgrade to a specific firmware load via its built-in web server interface.

Firmware upgrades are attempted only when the SPA-2000 is idle, since they trigger a software reboot.

Firmware Upgrade via PC Utility Program:
From time to time, Sipura Technology will make available a PC executable file that will facilitate the upgrade of a SPA-2000. In order to upgrade a device via this method, the end user must have administrative permission (via password protected log-in) to perform this upgrade.

Once the user has obtained the proper firmware upgrade executable, the user simply runs the program from a file location on their local PC. The PC program walks the user through the upgrade process via a graphical user interface. Generally, the entire upgrade process should take no more than five minutes to complete.

Please note: Some end-users who have obtained their SPA-2000 directly from a service provider will never need to manually upgrade their device. Via the remote upgrade process, Sipura Technology provides capability for the SPA-2000 to be maintained from a remote location (e.g. a service provider network server), using the Internet connection of the end-user as the conduit through which profile updates and firmware upgrades are performed.

3.2. IVR Interface

Administrators and/or users can check (read) and set (write) basic network configuration settings via a touchtone telephone connected to one of the RJ-11 phone ports of the SPA-2000.

Please Note: Service Providers offering service using the SPA-2000 may restrict, protect or turn off certain aspects of the unit's IVR and web configuration capabilities.

The Interactive Voice Response (IVR) capabilities of the SPA are designed to give the administrator and/or user basic read/write capabilities such that the unit can attain basic IP network connectivity and, if allowed by the service provider, the more advanced browser-based configuration menu may be accessed.

3.2.1. IVR Conventions:

1. The SPA IVR uses the following conventions: By factory default, there is no password. No password authentication is required for all the IVR settings. If Administrator password is set, password authentication will be prompted for certain IVR settings. See 3.3.2 for detailed information about administrator password.

Please note: The Administrator and User passwords may only be set via the SPA-2000 web interface.

To input the password using the phone keypad, the following translation convention applies:

- To input: A, B, C, a, b, c -- press '**2**'
- To input: D, E, F, d, e, f -- press '**3**'
- To input: G, H, I, g, h, i -- press '**4**'
- To input: J, K, L, j, k, l -- press '**5**'
- To input: M, N, O, m, n, o -- press '**6**'
- To input: P, Q, R, S, p, q, r, s -- press '**7**'
- To input: T, U, V, t, u, v -- press '**8**'

- To input: W, X, Y, Z, w, x, y, z -- press '**9**'
- To input all other characters in the administrator password, press '**0**'

Note: This translation convention only applies to the password input.

For example: to input password "**test#@1234**" by phone keypad, you need to press the following sequence of digits: **8378001234**.

2. After entering a value, press the **#** (pound) key to indicate end of input.

- To Save value, press '**1**'
- To Review the value, press '**2**'
- To Re-enter the value, press '**3**'
- To Cancel the value entry and return to the main configuration menu, press '*****' (star)

Please Note:

- The final '#' key will not be counted as part of the value.
- Saved settings will take effect when the telephone is hung-up and if necessary, the SPA-2000 will automatically reboot.

3. After one minute of inactivity, the unit times out. The user will need to re-enter the configuration menu from the beginning by pressing *** * * ***.

4. If, while entering a value (like an IP address), you decide to exit without entering any changes, you may do so by pressing the ***** (star) key **twice within a half second** window of time. Otherwise, the entry of the ***** (star) key will be treated as a dot (decimal point).

Example:

To enter an IP address, use numbers 0 – 9 on the telephone key pad. Use the * (star) key to enter a decimal point.

To enter the following IP address value: 192.168.2.215

A. Use the touchtone key pad to enter: **192*168*2*215#**

B. When prompted, enter **1** to save setting to configuration.

C. Hang-up the phone to cause setting to take effect.

- or -

D. Enter the value of the next setting category to modify . . .

5. Hang-up the phone to cause all settings to take effect.

3.2.2. SPA-2000 Interactive Voice Response (IVR) Menu:

IVR Action	IVR Menu Choice	Parameter(s)	Notes:
Enter IVR Menu	*** * * ***	None	Ignore SIT or other tones until you hear, "Sipura configuration menu. Please enter option followed by the pound key or hang-up to exit."
Check DHCP	**100**	None	IVR will announce if DHCP in enabled or disabled.

Enable/Disable DHCP	**101**	Enter 1 to enable Enter 0 to disable	
Check IP Address	**110**	None	IVR will announce the current IP address of SPA.
Set Static IP Address	**111**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	DHCP must be "Disabled" otherwise you will hear, "Invalid Option," if you try to set this value.
Check Network Mask	**120**	None	IVR will announce the current network mask of SPA.
Set Network Mask	**121**	Enter value using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	DHCP must be "Disabled" otherwise you will hear, "Invalid Option," if you try to set this value.
Check Static Gateway IP Address	**130**	None	IVR will announce the current gateway IP address of SPA.
Set Static Gateway IP Address	**131**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	DHCP must be "Disabled" otherwise you will hear, "Invalid Option," if you try to set this value.
Check MAC Address	**140**	None	IVR will announce the MAC address of SPA in hex string format.
Check Firmware Version	**150**	None	IVR will announce the version of the firmware running on the SPA.
Enable/Disable Web Server of SPA	**7932**	Enter 1 to enable Enter 0 to disable	**Requires Password**
Manual Reboot of Unit	**732668**	None	After you hear "Option Successful," hang-up. Unit will reboot automatically.
Factory Reset of Unit **WARNING:**	**73738**	Enter 1 to confirm Enter * (start) to cancel operation	SPA will prompt for confirmation. After confirming, you will hear "Option Successful." Hang-

ALL SETTINGS WILL BE LOST!			up. Unit will reboot and all configuration parameters will be reset to factory default values.

Note: If the Administrator password is not set, the items marked with "Requires Password" will not require a password.

3.3. SPA Web Interface

The SPA provides a built-in web server. Convenient configuration and administration can be performed through an integral web interface.

3.3.1. Web Interface Conventions

The SPA uses the following conventions with the web administration capabilities:

- The SPA-2000 web administration supports two privilege levels: Administrator and User. To use the User privilege, simply point a web browser at the IP address of the SPA-2000; to use the administrator privilege, use URL http://IP_Address_Of_SPA/admin/. See 3.3.2 for more information about administration privileges.
- Version 1.0 of the SPA supports Internet Explorer 5.5 and above and Netscape 7.0 and above.
- The web configuration pages can be password protected. See 3.3.2 for more information about password protect.
    - The user name of web Administrator is : **admin**
    - The user name of web User is : **user**
    - **Note**: The user names for both administrator and User are fixed and cannot be changed.
- After making changes to SPA-2000 configuration parameters, pressing "**Submit All Changes**" button will apply all the changes and if necessary, automatically reboot the device. Multiple changes may be made on multiple page tabs of the web interface at the same time. Pressing "**Submit All Changes**" will apply all the modifications.

  Important Note: switching between page tabs won't apply the changes to SPA-2000, The only way to apply the changes is to press the "**Submit All Changes**" button.
- If the "**Undo All Changes**" button is clicked, any modifications to profile parameters *on any and all pages* will be reset back to their original values before modification.

  NOTE: Pressing the "**Undo All Changes**" has no effect on the SPA-2000; it will only reset the values on the web page.

3.3.2. Administration Privileges

The SPA-2000 supports two levels of administration privileges: Administrator and User, both privileges can be password protected.

***Important note**: by factory default, there are no passwords assigned for both Administrator and User.*

The Administrator has the privilege to modify all web profile parameters and can also modify the passwords of both Administrator and User. A User only has the privilege to access part of the web

profile parameters. The parameter group that the User can access is specified by the Administrator, which can only be done through provisioning the SPA-2000 via the TFTP process.

To access the Administrator level privilege, use the following URL: http://IP_Address_Of_SPA/admin/. If the password has been set for Administrator, the browser will prompt for Administrator authentication. The username for Administrator is “admin” and cannot be changed.

To access the User level privilege, use URL: http://IP_Address_Of_SPA/. If the password has been set for User, the browser will prompt for User authentication. The username for User is “user” and cannot be changed.

When browsing Administrator pages, one can switch to User privileges by clicking the link “User Login”. (**Note**: if User password was set, the browser will prompt for User authentication when you click the “User Login” link). Conversely, from the User pages you can switch to Administrator privilege by clicking the link “Admin Login.” Authentication is needed if the Administrator password has been previously set.

Warning: *Switching between the User and Administrator will discard the uncommitted changes that have already been made on the web pages.*

3.3.3. Basic and Advanced Views

The web configuration interface provides a Basic and an Advanced view from which the various configuration parameters can be accessed. The SPA Provisioning tab is only visible from the Advanced Administrator view of the web interface.

Warning: Switching between the basic and advanced view will discard the uncommitted changes that have already been made on the web pages.

3.3.4. SPA-2000 Web Page Configuration Default Values

The following figures depict the default values and web page locations of the SPA-2000 web administration and configuration pages. The SPA-2000 Administrator Log-in, Advanced screen settings are shown below.

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SPA-2000 Configuration

Info | System | SIP | Provisioning | Regional | Line 1 | Line 2 | User 1 | User 2 | User Login basic | advanced

System Information

DHCP:	Yes	Current IP:	192.168.2.213
Host Name:		Domain:	sipura.com
Current Netmask:	255.255.255.0	Current Gateway:	192.168.2.1
Primary DNS:	192.168.2.2		
Secondary DNSs:	192.168.2.1		

Product Information

Product Name:	SPA-2000	Serial Number:	88012BZZ000028
Software Version:	1.0.7	Hardware Version:	0.99.1
MAC Address:	000E08000028		

System Status

Current Time:	11/10/2003 11:49:16	Elapsed Time:	00:31:08
Broadcast Pkts Sent:	3	Broadcast Bytes Sent:	1026
Broadcast Pkts Recv:	428	Broadcast Bytes Recv:	49583
Broadcast Pkts Dropped:	0	Broadcast Bytes Dropped:	0
RTP Packets Sent:	0	RTP Bytes Sent:	0
RTP Packets Recv:	0	RTP Bytes Recv:	0
SIP Messages Sent:	227	SIP Bytes Sent:	54527
SIP Messages Recv:	235	SIP Bytes Recv:	73011
External IP:			

Line 1 Status

Hook State:	On	Registration State:	Registered
Last Registration At:	11/10/2003 11:47:54	Next Registration In:	812 s
Message Waiting:	No	Call Back Active:	No
Last Called Number:		Last Caller Number:	
Mapped SIP Port:			
Call 1 State:	Idle	Call 2 State:	Idle
Call 1 Tone:	None	Call 2 Tone:	None
Call 1 Encoder:		Call 2 Encoder:	
Call 1 Decoder:		Call 2 Decoder:	
Call 1 FAX:		Call 2 FAX:	
Call 1 Type:		Call 2 Type:	
Call 1 Remote Hold:		Call 2 Remote Hold:	
Call 1 Callback:		Call 2 Callback:	
Call 1 Peer Name:		Call 2 Peer Name:	
Call 1 Peer Phone:		Call 2 Peer Phone:	
Call 1 Duration:		Call 2 Duration:	
Call 1 Packets Sent:		Call 2 Packets Sent:	
Call 1 Packets Recv:		Call 2 Packets Recv:	
Call 1 Bytes Sent:		Call 2 Bytes Sent:	
Call 1 Bytes Recv:		Call 2 Bytes Recv:	
Call 1 Decode Latency:		Call 2 Decode Latency:	
Call 1 Jitter:		Call 2 Jitter:	
Call 1 Frames Lost:		Call 2 Frames Lost:	
Call 1 Packet Error:		Call 2 Packet Error:	
Call 1 Mapped RTP Port:		Call 2 Mapped RTP Port:	

Line 2 Status

Hook State:	On	Registration State:	Registered
Last Registration At:	11/10/2003 11:18:08	Next Registration In:	1704 s
Message Waiting:	No	Call Back Active:	No
Last Called Number:		Last Caller Number:	
Mapped SIP Port:			
Call 1 State:	Idle	Call 2 State:	Idle
Call 1 Tone:	None	Call 2 Tone:	None
Call 1 Type:		Call 2 Type:	
Call 1 Encoder:		Call 2 Encoder:	
Call 1 Decoder:		Call 2 Decoder:	
Call 1 FAX:		Call 2 FAX:	
Call 1 Remote Hold:		Call 2 Remote Hold:	
Call 1 Callback:		Call 2 Callback:	
Call 1 Peer Name:		Call 2 Peer Name:	
Call 1 Peer Phone:		Call 2 Peer Phone:	
Call 1 Duration:		Call 2 Duration:	
Call 1 Packets Sent:		Call 2 Packets Sent:	
Call 1 Packets Recv:		Call 2 Packets Recv:	
Call 1 Bytes Sent:		Call 2 Bytes Sent:	
Call 1 Bytes Recv:		Call 2 Bytes Recv:	
Call 1 Decode Latency:		Call 2 Decode Latency:	
Call 1 Jitter:		Call 2 Jitter:	
Call 1 Frames Lost:		Call 2 Frames Lost:	
Call 1 Packet Error:		Call 2 Packet Error:	
Call 1 Mapped RTP Port:		Call 2 Mapped RTP Port:	

Undo All Changes | Submit All Changes

User Login basic | advanced

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Figure 1 – SPA-2000 Information Web Page (Advanced Admin View)

Figure 2 – SPA-2000 System Configuration Web Page (Advanced Admin View)

Figure 3 – SPA-2000 SIP Configuration Web Page (Advanced Admin View)

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SPA-2000 Configuration
Info | System | SIP | Provisioning | Regional | Line 1 | Line 2 | User 1 | User 2
User Login basic | advanced
Configuration Profile
Provision Enable: no
Resync On Reset: yes
Resync Random Delay: 60
Resync Periodic: 3600
Resync Error Retry Delay: 3600
Resync From SIP: yes
Profile Rule:
Log Resync Request Msg: $(PN) $(MAC) -- Requesting resync $(METH)://$(SERVIP):$(PORT)$(PATH)
Log Resync Success Msg: $(PN) $(MAC) -- Successful resync $(METH)://$(SERVIP):$(PORT)$(PATH)
Log Resync Failure Msg: $(PN) $(MAC) -- Resync failed: $(ERR)
Firmware Upgrade
Upgrade Enable: yes
Upgrade Error Retry Delay: 86400
Upgrade Rule: (<0.99.52)?tftp://192.168.2.251/spa.bin
Log Upgrade Request Msg: $(PN) $(MAC) -- Requesting upgrade $(METH)://$(SERVIP):$(PORT)$(PATH)
Log Upgrade Success Msg: $(PN) $(MAC) -- Successful upgrade $(METH)://$(SERVIP):$(PORT)$(PATH) -- $
Log Upgrade Failure Msg: $(PN) $(MAC) -- Upgrade failed: $(ERR)
General Purpose Parameters
GPP_A:
GPP_B:
GPP_C:
GPP_D:
Undo All Changes
Submit All Changes
User Login basic | advanced
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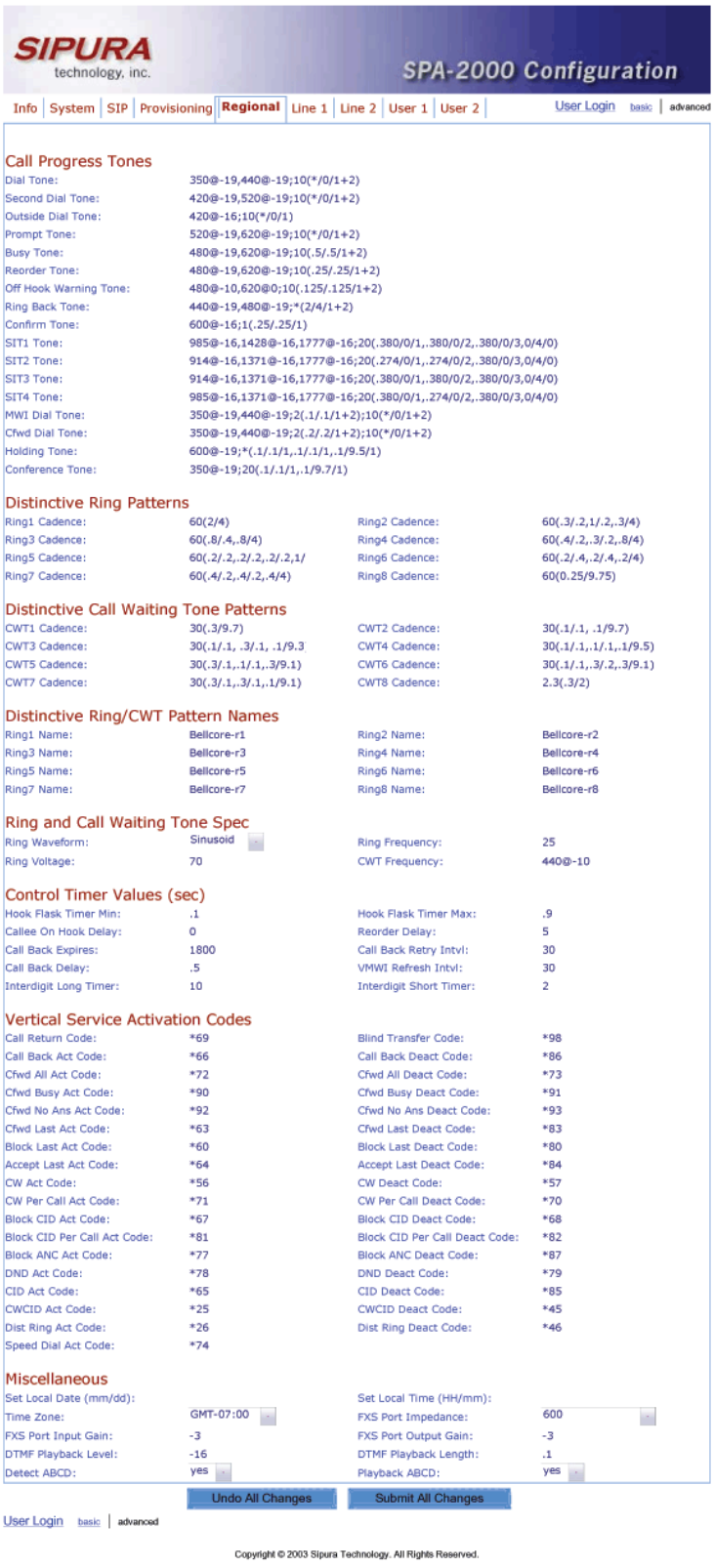
Figure 4 – SPA-2000 Provisioning Configuration Web Page (Advanced Admin View)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SPA-2000 Configuration
Info | System | SIP | Provisioning | Regional | Line 1 | Line 2 | User 1 | User 2
User Login basic | advanced
Call Progress Tones
Dial Tone: 350@-19,440@-19;10(*/0/1+2)
Second Dial Tone: 420@-19,520@-19;10(*/0/1+2)
Outside Dial Tone: 420@-16;10(*/0/1)
Prompt Tone: 520@-19,620@-19;10(*/0/1+2)
Busy Tone: 480@-19,620@-19;10(.5/.5/1+2)
Reorder Tone: 480@-19,620@-19;10(.25/.25/1+2)
Off Hook Warning Tone: 480@-10,620@0;10(.125/.125/1+2)
Ring Back Tone: 440@-19,480@-19;*(2/4/1+2)
Confirm Tone: 600@-16;1(.25/.25/1)
SIT1 Tone: 985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT2 Tone: 914@-16,1371@-16,1777@-16;20(.274/0/1,.274/0/2,.380/0/3,0/4/0)
SIT3 Tone: 914@-16,1371@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT4 Tone: 985@-16,1371@-16,1777@-16;20(.380/0/1,.274/0/2,.380/0/3,0/4/0)
MWI Dial Tone: 350@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2)
Cfwd Dial Tone: 350@-19,440@-19;2(.2/.2/1+2);10(*/0/1+2)
Holding Tone: 600@-19;*(.1/.1/1,.1/.1/1,.1/9.5/1)
Conference Tone: 350@-19;20(.1/.1/1,.1/9.7/1)
Distinctive Ring Patterns
Ring1 Cadence: 60(2/4)
Ring2 Cadence: 60(.3/.2,1/.2,.3/4)
Ring3 Cadence: 60(.8/.4,.8/4)
Ring4 Cadence: 60(.4/.2,.3/.2,.8/4)
Ring5 Cadence: 60(.2/.2,.2/.2,.2/.2,1/
Ring6 Cadence: 60(.2/.4,.2/.4,.2/4)
Ring7 Cadence: 60(.4/.2,.4/.2,.4/4)
Ring8 Cadence: 60(0.25/9.75)
Distinctive Call Waiting Tone Patterns
CWT1 Cadence: 30(.3/9.7)
CWT2 Cadence: 30(.1/.1, .1/9.7)
CWT3 Cadence: 30(.1/.1, .3/.1, .1/9.3
CWT4 Cadence: 30(.1/.1,.1/.1,.1/9.5)
CWT5 Cadence: 30(.3/.1,.1/.1,.3/9.1)
CWT6 Cadence: 30(.1/.1,.3/.2,.3/9.1)
CWT7 Cadence: 30(.3/.1,.3/.1,.1/9.1)
CWT8 Cadence: 2.3(.3/2)
Distinctive Ring/CWT Pattern Names
Ring1 Name: Bellcore-r1
Ring2 Name: Bellcore-r2
Ring3 Name: Bellcore-r3
Ring4 Name: Bellcore-r4
Ring5 Name: Bellcore-r5
Ring6 Name: Bellcore-r6
Ring7 Name: Bellcore-r7
Ring8 Name: Bellcore-r8
Ring and Call Waiting Tone Spec
Ring Waveform: Sinusoid
Ring Frequency: 25
Ring Voltage: 70
CWT Frequency: 440@-10
Control Timer Values (sec)
Hook Flask Timer Min: .1
Hook Flask Timer Max: .9
Callee On Hook Delay: 0
Reorder Delay: 5
Call Back Expires: 1800
Call Back Retry Intvl: 30
Call Back Delay: .5
VMWI Refresh Intvl: 30
Interdigit Long Timer: 10
Interdigit Short Timer: 2
Vertical Service Activation Codes
Call Return Code: *69
Blind Transfer Code: *98
Call Back Act Code: *66
Call Back Deact Code: *86
Cfwd All Act Code: *72
Cfwd All Deact Code: *73
Cfwd Busy Act Code: *90
Cfwd Busy Deact Code: *91
Cfwd No Ans Act Code: *92
Cfwd No Ans Deact Code: *93
Cfwd Last Act Code: *63
Cfwd Last Deact Code: *83
Block Last Act Code: *60
Block Last Deact Code: *80
Accept Last Act Code: *64
Accept Last Deact Code: *84
CW Act Code: *56
CW Deact Code: *57
CW Per Call Act Code: *71
CW Per Call Deact Code: *70
Block CID Act Code: *67
Block CID Deact Code: *68
Block CID Per Call Act Code: *81
Block CID Per Call Deact Code: *82
Block ANC Act Code: *77
Block ANC Deact Code: *87
DND Act Code: *78
DND Deact Code: *79
CID Act Code: *65
CID Deact Code: *85
CWCID Act Code: *25
CWCID Deact Code: *45
Dist Ring Act Code: *26
Dist Ring Deact Code: *46
Speed Dial Act Code: *74
Miscellaneous
Set Local Date (mm/dd):
Set Local Time (HH/mm):
Time Zone: GMT-07:00
FXS Port Impedance: 600
FXS Port Input Gain: -3
FXS Port Output Gain: -3
DTMF Playback Level: -16
DTMF Playback Length: .1
Detect ABCD: yes
Playback ABCD: yes
Undo All Changes
Submit All Changes
User Login basic | advanced
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Figure 5 – SPA-2000 Regional Configuration Web Page (Advanced Admin View)

SIPURA technology, inc.
SPA-2000 Configuration
Info | System | SIP | Provisioning | Regional | Line 1 | Line 2 | User 1 | User 2
User Login basic | advanced
Line Enable: yes
SIP Port: 5060
EXT SIP Port:
SIP TOS/DiffServ Value: 0x68
RTP TOS/DiffServ Value: 0xb8
NAT Mapping Enable: no
NAT Keep Alive Enable: no
NAT Keep Alive Msg: $NOTIFY
NAT Keep Alive Dest: $PROXY
SIP Debug Option: full
Proxy and Registration
Proxy: preproxy2.vonage.net
Outbound Proxy:
Register: yes
Register Expires: 900
Use DNS SRV: no
Use Outbound Proxy: no
Use OB Proxy In Dialog: yes
Make Call Without Reg: no
Ans Call Without Reg: no
Subscriber Information
Display Name:
Password: **********
Auth ID:
User ID: 14088501233
Use Auth ID: no
Supplementary Service Subscription
Call Waiting Serv: yes
Block ANC Serv: yes
Cfwd All Serv: yes
Cfwd No Ans Serv: yes
Cfwd Last Serv: yes
Accept Last Serv: yes
CID Serv: yes
Call Return Serv: yes
Three Way Call Serv: yes
Attn Transfer Serv: yes
MWI Serv: yes
Speed Dial Serv: yes
Block CID Serv: yes
Dist Ring Serv: yes
Cfwd Busy Serv: yes
Cfwd Sel Serv: yes
Block Last Serv: yes
DND Serv: yes
CWCID Serv: yes
Call Back Serv: yes
Three Way Conf Serv: yes
Unattn Transfer Serv: yes
VMWI Serv: yes
Audio Configuration
Preferred Codec: G711u
Use Pref Codec Only: yes
Silence Supp Enable: yes
Echo Canc Adapt Enable: yes
FAX Detect Enable: yes
FAX Passthru Codec: G711u
DTMF Tx Method: Auto
LBR Codec Enable: yes
Echo Canc Enable: yes
Echo Supp Enable: yes
FAX Passthru Enable: yes
FAX Codec Symmetric: yes
Dial Plan
Dial Plan: (*xx|011,xx.|[3469]11|0|00|<:1408>[2-9]xxxxxx|1[2-9]xx[2-9]xxxxxx|xxxxx)
Enable IP Dialing: no
FXS Port Polarity Configuration
Idle Polarity: Forward
Callee Conn Polarity: Reverse
Caller Conn Polarity: Reverse
Undo All Changes
Submit All Changes
User Login basic | advanced
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Figure 6 – SPA-2000 Line 1 Configuration Web Page (Line 2 Is Identical)

Figure 7 – SPA-2000 User 1 Configuration Web Page (User 2 Is Identical)

3.4. Configuration Parameters

3.4.1. System Parameters

System Configuration

Parameter Name	Description	Default
Admin Password	The password for administrator	
User Password	The password for User	
Enable Web Server	Enable/disable web server of SPA **This feature should only be used on firmware version 1.0.9 or later.**	Yes

Network Configuration

Parameter Name	Description	Default
DHCP	Enable/Disable DHCP	Yes
Host Name	Host Name of SPA	
Domain Name	The network domain of SPA	
Static IP	Static IP address of SPA, which will take effect if DHCP is disabled	0.0.0.0
NetMask	The NetMask used by SPA when DHCP is disabled	255.255.255.0
Gateway	The default gateway used by SPA when DHCP is disabled	0.0.0.0
Primary DNS	DNS server used by SPA in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this will be the primary DNS server.	0.0.0.0
Secondary DNS	DNS server used by SPA in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this will be the secondary DNS server.	0.0.0.0
DNS Query Mode	Do parallel or sequential DNS Query	Parallel
Syslog Server	Specify the syslog server name and port. This feature specifies the server for logging SPA system information and critical events.	
Debug Server	The debug server name and port. This feature specifies the server for logging SPA debug information. The level of detailed output depends on the debug level parameter setting.	
Debug Level	The higher the debug level, the more debug information will be generated. Zero (0) means no debug information will be generated.	0

Notes:
- Parallel DNS query mode: SPA-2000 will send the same request to all the DNS servers at the same time when doing a DNS lookup, the first incoming reply will be accepted by SPA-2000.
- To log SIP messages, Debug Level must be set to at least 2.
- If both Debug Server and Syslog Server are specified, syslog messages are also logged to the Debug Server.

3.4.2. Provisioning Parameters

Provisioning operations are gated by the Provision_Enable parameter.

Parameter Name	Description	Default
Provision Enable	Master enable for configuration profile resync operations	Yes

Resync On Reset	Resyncs configuration profile from configuration server whenever the SPA-2000 resets.	Yes
Resync Random Delay	Spread interval for resync requests	2
Resync Periodic	Resyncs configuration profile periodically after reset.	3600
Resync Error Retry Delay	Retry interval following resync failure	3600
Resync From SIP	Enables resync of configuration profile from a SIP command.	Yes
Profile Rule	Configuration profile URL script.	/spa.cfg
Log Resync Request Msg	Syslog message generated when attempting a resync	
Log Resync Success Msg	Syslog message generated after a successful resync	
Log Resync Failure Msg	Syslog message generated after a failed resync	
GPP A	General purpose parameter	empty
GPP B	General purpose parameter	empty
GPP C	General purpose parameter	empty
GPP D	General purpose parameter	empty

Note: In a customized SPA-2000, the profile rule would point to a service provider's server.

3.4.3. Upgrade Parameters

Parameter Name	Description	Default
Upgrade Enable	Master enable for firmware upgrade operations	Yes
Upgrade Error Retry Delay	Retry interval following upgrade failure	3600
Upgrade Rule	Upgrade script.	empty
Log Upgrade Request Msg	Syslog message generated when attempting an upgrade	
Log Upgrade Success Msg	Syslog message generated after a successful upgrade	
Log Upgrade Failure Msg	Syslog message generated after a failed upgrade	

Note: In a customized SPA-2000, the upgrade rule would point to a service provider's server.

3.4.4. Protocol Parameters

Parameter Name	Description	Default
Max Forward	SIP Max-Forward value. Range: 1 – 255	70
Max Redirection	Number of times to allow an INVITE to be redirected by a 3xx response to avoid an infinite loop. **Note: This parameter currently has no effect: there is no limit on number of redirection.**	5
Max Auth	Maximum number of times a request may be challenged (0-255)	2
SIP User Agent Name	User-Agent Header to be used by the unit in outbound requests. If empty, the header is not included.	Sipura/ $version
SIP Server Name	Server Header to used by the unit in responses to inbound responses. If empty, the header is not included.	Sipura/ $version

SIP Accept Language	Accept-Language Header to be used by the unit. If empty, the header is not included.	
Remove Last Reg	Remove last registration before registering a new one if value is different one.	no
SIP T1	RFC 3261 T1 value (RTT Estimate). Range: 0 – 64 sec	.5
SIP T2	RFC 3261 T2 value (Maximum retransmit interval for non-INVITE requests and INVITE responses). Range: 0 – 64 sec	4
SIP T4	RFC 3261 T4 value (Maximum duration a message will remain in the network). Range: 0 – 64 sec	5
SIP Timer B	INVITE time out value. Range: 0 – 64 sec	32
SIP Timer F	Non-INVITE time out value. Range: 0 – 64 sec	32
SIP Timer H	INVITE final response time out value. Range: 0 – 64 sec	32
SIP Timer D	ACK hang around time. Range: 0 – 64 sec	32
SIP Timer J	Non-INVITE response hang around time. Range: 0 – 64 sec	32
INVITE Expires	INVITE request Expires header value in sec. 0 = do not include Expires header in INVITE. Range: 0 – (2^{31} – 1)	180
ReINVITE Expires	ReINVITE request Expires header value in sec. 0 = do not include Expires header in the request. Range: 0 – (2^{31} – 1)	30
Reg Min Expires	Minimum registration expiration time allowed from the proxy in the Expires header or as a Contact header parameter. If proxy returns something less this value, then the minimum value is used.	1
Reg Max Expires	Maximum registration expiration time allowed from the proxy in the Min-Expires header. If value is larger than this, then the maximum value is used	7200
Reg Retry Intvl	Interval to wait before the SPA retries registration again after encountering a failure condition during last registration	30
SIT1 RSC[1]	SIP response status code to INVITE on which to play the SIT1 Tone	
SIT2 RSC[1]	SIP response status code to INVITE on which to play the SIT2 Tone	
SIT3 RSC[1]	SIP response status code to INVITE on which to play the SIT3 Tone	
SIT4 RSC[1]	SIP response status code to INVITE on which to play the SIT4 Tone	
Try Backup RSC	SIP response status code on which to retry a backup server for the current request	
Retry Reg RSC	Interval to wait before the SPA retries registration again after encountering a failure condition during last registration	30
RTP Port Min[2]	Minimum port number for RTP transmission and reception	16384
RTP Port Max[2]	Maximum port number for RTP transmission and reception	16482
RTP Packet Size	Packet size in sec. Valid values must be multiple of 0.01s. Range: 0.01 – 0.16	0.02

Notes:
1. Reorder or Busy Tone will be played by default for all unsuccessful response status code
2. <RTP Port Min> and <RTP Port Max> should define a range that contains at least 4 even number ports, such as 100 – 106

3.4.4.1. Dynamic Payload Types

Parameter Name	Description	Default

NSE Dynamic Payload[1,2]	NSE dynamic payload type	100
AVT Dynamic Payload[1,2]	AVT dynamic payload type	101
G726r16 Dynamic Payload[1,2]	G726-16 dynamic payload type	98
G726r24 Dynamic Payload[1,2]	G726-24 dynamic payload type	97
G726r40 Dynamic Payload[1,2]	G726-40 dynamic payload type	96
G729b Dynamic Payload[1,2]	G729b dynamic payload type	99

Notes:

1. Valid range is 96 – 127

2. The configured dynamic payloads are used for outbound calls only where the SPA-2000 presents the SDP offer. For inbound calls with a SDP offer, SPA-2000 will follow the caller's dynamic payload type assignments

3.4.4.2. SDP Audio Codec Names

Parameter Name	Description	Default
NSE Codec Name	NSE Codec name used in SDP	NSE
AVT Codec Name	AVT Codec name used in SDP	telephone-event
G711a Codec Name	G711a Codec name used in SDP	PCMA
G711u Codec Name	G711u Codec name used in SDP	PCMU
G726r16 Codec Name	G726-16 Codec name used in SDP	G726-16
G726r24 Codec Name	G726-24 Codec name used in SDP	G726-24
G726r32 Codec Name	G726-32 Codec name used in SDP	G726-32
G726r40 Codec Name	G726-40 Codec name used in SDP	G726-40
G729a Codec Name	G729a Codec name used in SDP	G729a
G729b Codec Name	G729b Codec name used in SDP	G729ab

Notes:

1. SPA-2000 uses the configured codec names in its outbound SDP

2. SPA-2000 ignores the codec names in incoming SDP for standard payload types (0 – 95).

3. For dynamic payload types, SPA-2000 identifies the codec by the configured codec names. Comparison is case-insensitive.

3.4.4.3. NAT Support

Parameter Name	Description	Default
Handle_VIA_received	If set to "yes", the SPA will process the "received" parameter in the VIA header inserted by the server in a response to any one of its request. Else the parameter is ignored.	No
Handle_VIA_rport	If set to "yes", the SPA will process the "rport" parameter in the VIA header inserted by the server in a response to any one of its request. Else the parameter is ignored.	No
Insert VIA received	Insert received parameter in VIA header in SIP responses if received from IP and VIA sent-by IP differ	No
Insert VIA rport	Insert rport parameter in VIA header in SIP responses if received-from port and VIA sent-by port differ	No
Substitute VIA addr	Use nat-mapped IP:port values in VIA header	No
Send Resp To Src Port	Send response to the request source port instead of the VIA sent-by port	No
STUN Server	STUN server to contact for NAT mapping discovery	
STUN Enable	Enable the use of STUN to discover NAT mapping	No

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Ext IP	External IP address to substitute for the actual IP address of the unit in all outgoing SIP messages. If “0.0.0.0” is specified, no IP address substitution is performed.	0.0.0.0
Ext RTP Port Min	External port mapping of <RTP Port Min>. If this value is non-zero, the RTP port number in all outgoing SIP messages is substituted by the corresponding port value in the external RTP port range.	0
NAT Keep Alive Intvl	Interval between sending NAT-mapping keep alive message in sec	15

Notes:

3.4.5. Line 1 and Line 2 Parameters

Per line parameter tags must be appended with [1] or [2] (corresponding to lines 1 or 2) in the configuration profile. It is omitted below for readability.

3.4.5.1. User Account Information

Parameter Name	Description	Default
Line Enable	Enable this line for service	Yes
SIP Port	SIP message listening port and transmission port	5060
Ext SIP Port	External port to substitute for the actual SIP port of the unit in all outgoing SIP messages. If “0” is specified, no SIP port substitution is performed.	0
SIP TOS/DiffServ Value	TOS/DiffServ field value in UDP IP Packets carrying a SIP Message	0x68
RTP TOS/DiffServ Value	TOS/DiffServ field value in UDP IP Packets carrying a RTP data	0xb8
NAT Mapping Enable	Enable the use of externally mapped of IP address and SIP/RTP ports in SIP messages. The mapping may be discovered by any of the supported methods.	No
NAT Keep Alive Enable	If set to “yes”, the configured <NAT Keep Alive Msg> is sent periodically every <NAT Keep Alive Intvl> seconds.	No
NAT Keep Alive Msg	Contents of the keep-alive message to be sent to a given destination periodically to maintain the current NAT-mapping. It could be an empty string. If value is $NOTIFY, a NOTIFY message is sent as keep alive. If value is $REGISTER, a REGISTER message w/o Contact is sent.	$NOTIFY
NAT Keep Alive Dest	Destination to send NAT keep alive messages to. If value is $PROXY, it will be sent to the current proxy or outbound proxy	$PROXY
SIP Debug Option	None, 1-line, full, exclude OPTIONS, exclude REGISTER, exclude NOTIFY, …	none
Proxy	SIP Proxy Server for all outbound requests	
Use Outbound Proxy	Enable the use of <Outbound Proxy>. If set to “no”, <Outbound Proxy> and <Use OB Proxy in Dialog) is ignored.	No
Outbound Proxy	SIP Outbound Proxy Server where all outbound requests are sent as the first hop.	No

Use OB Proxy In Dialog	Whether to forcer SIP requests to be sent to the outbound proxy within a dialog. Ignored if <Use Outbound Proxy> is “no” or <Outbound Proxy> is empty	Yes
Register	Enable periodic registration with the <Proxy>. This parameter is ignored if <Proxy> is not specified.	Yes
Make Call Without Reg	Allow making outbound calls without successful (dynamic) registration by the unit. If “No”, dial tone will not play unless registration is successful	No
Ans Call Without Reg	Allow answering inbound calls without successful (dynamic) registration by the unit	No
Register Expires[1]	Expires value in sec in a REGISTER request. SPA will periodically renew registration shortly before the current registration expired. This parameter is ignored if <Register> is “no”. Range: 0 – (2^{31} – 1) sec	3600
Use DNS SRV	Whether to use DNS SRV lookup for Proxy and Outbound Proxy	No
Display Name	Subscriber’s display name to appear in caller-id	
User ID	Subscriber’s user-id. Usually a E.164 number	
Password	Subscriber’s a/c password	
Auth ID	Subscriber’s authentication ID	
Use Auth ID	If set to “yes”, the pair <Auth ID> and <Password> are used for SIP authentication. Else the pair <User ID> and <Password> are used.	No

Notes:

1. If proxy responded to REGISTER with a smaller Expires value, the SPA will renew registration based on this smaller value instead of the configured value. If registration failed with an “Expires too brief” error response, the SPA will retry with the value given in the Min-Expires header in the error response.

3.4.5.2. Supplementary Services Enable

The SPA-2000 provides native support of a large set of enhanced or supplementary services. All of these services are optional. The parameters listed in the following table are used to enable or disable a specific supplementary service. A supplementary service should be disabled if a) the user has not subscribed for it, or b) the Service Provider intends to support similar service using other means than relying on the SPA-2000.

Parameter Name	Description	Default
CW Serv	Enable Call Waiting Service	Yes
Block CID Serv	Enable Block Caller ID Service	Yes
Block ANC Serv	Enable Block Anonymous Calls Service	Yes
Dist Ring Serv	Enable Distinctive Ringing Service	Yes
Cfwd All Serv	Enable Call Forward All Service	Yes
Cfwd Busy Serv	Enable Call Forward Busy Service	Yes
Cfwd No Ans Serv	Enable Call Forward No Answer Service	Yes
Cfwd Sel Serv	Enable Call Forward Selective Service	Yes
DND Serv	Enable Do Not Disturb Service	Yes
CID_Serv	Enable Caller ID Service	Yes
CWCID Serv	Enable Call Waiting Caller ID Service	Yes

Call Return Serv	Enable Call Return Service	Yes
Call Back Serv	Enable Call Back Service	Yes
Three Way Call Serv[1]	Enable Three Way Calling Service	Yes
Three Way Conf Serv[1,2]	Enable Three Way Conference Service	Yes
Attn Transfer Serv[1,2]	Enable Attended Call Transfer Service	Yes
Unattn Transfer Serv	Enable Unattended (Blind) Call Transfer Service	Yes
MWI Serv[3]	Enable MWI Service	Yes
VMWI Serv	Enable VMWI Service (FSK)	Yes
Block Last Serv	Enable Block Last Call Service	Yes
Accept Last Serv	Enable Accept Last Call Service	Yes
Cfwd Last Serv	Enable Forward Last Call Service	Yes
Speed Dial Serv	Enable Speed Dial Service	Yes

Notes:

1. Three Way Calling is required for Three Way Conference and Attended Transfer.
2. Three Way Conference is required for Attended Transfer.
3. MWI is available only if a Voice Mail Service is set-up in the deployment.

3.4.5.3. Audio Settings

Parameter Name	Description	Default
Preferred Codec	Select a preferred codec for all calls. However, the actual codec used in a call still depends on the outcome of the codec negotiation protocol.	G711u
Use Pref Codec Only	Only use the preferred codec for all calls. The call will fail if the far end does not support this codec.	No
LBR Codec Enable	Enable of the use of Low Bit Rate Codec, such as G726 and G729. If set to “no”, only G711u or G711a codec will be used in all calls on this line	Yes
Silence Supp Enable	Enable silence suppression so that silent audio frames are not transmitted	No
Echo Canc Enable	Enable the use of echo canceller	Yes
Echo Canc Adapt Enable	Enable echo canceller to adapt	Yes
Echo Supp Enable	Enable the use of echo suppressor. If <Echo Canc Enable> is “no”, this parameter is ignored	Yes
FAX Detect Enable	Enable detection of FAX tone. Ignored if <FAX Passthru Enable> is “no”	Yes
FAX Passthru Enable	Enable FAX pass-through support	Yes
DTMF Tx Method	Method to transmit DTMF signals to the far end: Inband = Send DTMF using the audio path; INFO = Use the SIP INFO method, AVT = Send DTMF as AVT events; Auto = Use In-band or AVT (RFC 2833) based on outcome of codec negotiation	Auto
FAX Passthru Codec	Codec to use for fax pass-through	G711u
FAX Codec Symmetric	Force unit to use symmetric codec during FAX pass-through	yes

Notes:

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3.4.5.4. Dial Plan

Parameter Name	Description	Default
Dial Plan	Per-line dial plan script	See below
Enable IP Dialing	Enable IP Dialing	no

Default Dial Plan script for each line:

"(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxx|xxxxxxxxxxxx.)"

Explanation of Default Dial Plan:

Dial Plan Entry	Functionality
*xx	Allow arbitrary 2 digit star code
[3469]11	Allow x11 sequences
0	Operator
00	Int'l Operator
[2-9]xxxxxx	US "local" number
1xxx[2-9]xxxxxx	US 1 + 10-digit long distance number
xxxxxxxxxxxx.	Everything else (Int'l long distance, FWD, ...)

Note: If IP dialing is enabled, one can dial [user-id@]a.b.c.d[:port], where '@', '.', and ':' are dialed by entering "*", user-id must be numeric (like a phone number) and a, b, c, d must be between 0 and 255, and port must be larger than 255. If port is not given, 5060 is used. Port and User-Id are optional. If the user-id portion matches a pattern in the dial plan, then it is interpreted as a regular phone number according to the dial plan. The INVITE message, however, is still sent to the outbound proxy if it is enabled.

3.4.5.5. Polarity Settings

Parameter Name	Description	Default
Idle Polarity	Polarity before call connected	Forward
Caller Conn Polarity	Polarity after outbound call connected	Reverse
Callee Conn Polarity	Polarity after inbound call connected	Reverse

Notes:

3.4.6. User 1 and User 2 Parameters

User 1/2 refers to the subscriber of Line 1/2. When a call is made from Line 1/2, SPA shall use the user and line settings for that Line; there is no user login support in SPA v1.0. Per user parameter tags must be appended with [1] or [2] (corresponding to line 1 or 2) in the configuration profile. It is omitted below for readability.

3.4.6.1. Call Forward And Selective Call Forward/Blocking Settings

Parameter Name	Description	Default

Cfwd All Dest	Forward number for Call Forward All Service	
Cfwd Busy Dest	Forward number for Call Forward Busy Service	
Cfwd No Ans Dest	Forward number for Call Forward No Answer Service	
Cfwd No Ans Delay	Delay in sec before Call Forward No Answer triggers	20
Cfwd Sel1 Caller	Caller number pattern to trigger Call Forward Selective 1	
Cfwd Sel2 Caller	Caller number pattern to trigger Call Forward Selective 2	
Cfwd Sel3 Caller	Caller number pattern to trigger Call Forward Selective 3	
Cfwd Sel4 Caller	Caller number pattern to trigger Call Forward Selective 4	
Cfwd Sel5 Caller	Caller number pattern to trigger Call Forward Selective 5	
Cfwd Sel6 Caller	Caller number pattern to trigger Call Forward Selective 6	
Cfwd Sel7 Caller	Caller number pattern to trigger Call Forward Selective 7	
Cfwd Sel8 Caller	Caller number pattern to trigger Call Forward Selective 8	
Cfwd Sel1 Dest	Forward number for Call Forward Selective 1	
Cfwd Sel2 Dest	Forward number for Call Forward Selective 2	
Cfwd Sel3 Dest	Forward number for Call Forward Selective 3	
Cfwd Sel4 Dest	Forward number for Call Forward Selective 4	
Cfwd Sel5 Dest	Forward number for Call Forward Selective 5	
Cfwd Sel6 Dest	Forward number for Call Forward Selective 6	
Cfwd Sel7 Dest	Forward number for Call Forward Selective 7	
Cfwd Sel8 Dest	Forward number for Call Forward Selective 8	
Block Last Caller	ID of caller blocked via the “Block Last Caller” service	
Accept Last Caller	ID of caller accepted via the “Accept Last Caller” service	

3.4.6.2. Speed Dial Settings

Parameter Name	Description	Default
Speed Dial 2	Target phone number (or URL) assigned to speed dial “2”	
Speed Dial 3	Target phone number (or URL) assigned to speed dial “3”	
Speed Dial 4	Target phone number (or URL) assigned to speed dial “4”	
Speed Dial 5	Target phone number (or URL) assigned to speed dial “5”	
Speed Dial 6	Target phone number (or URL) assigned to speed dial “6”	
Speed Dial 7	Target phone number (or URL) assigned to speed dial “7”	
Speed Dial 8	Target phone number (or URL) assigned to speed dial “8”	
Speed Dial 9	Target phone number (or URL) assigned to speed dial “9”	

3.4.6.3. Supplementary Service Settings

Parameter Name	Description	Default
CW Setting	Call Waiting on/off for all calls	Yes
Block CID Setting	Block Caller ID on/off for all calls	No
Block ANC Setting	Block Anonymous Calls on or off	No
DND Setting	DND on or off	No
CID Setting	Caller ID Generation on or off	Yes
CWCID Setting	Call Waiting Caller ID Generation on or off	Yes
Dist Ring Setting	Distinctive Ring on or off	Yes

3.4.6.4. Distinctive Ring and Ring Settings

Parameter Name	Description	Default

Ring 1 Caller	Caller number pattern to play Distinctive Ring/CWT 1	
Ring 2 Caller	Caller number pattern to play Distinctive Ring/CWT 2	
Ring 3 Caller	Caller number pattern to play Distinctive Ring/CWT 3	
Ring 4 Caller	Caller number pattern to play Distinctive Ring/CWT 4	
Ring 5 Caller	Caller number pattern to play Distinctive Ring/CWT 5	
Ring 6 Caller	Caller number pattern to play Distinctive Ring/CWT 6	
Ring 7 Caller	Caller number pattern to play Distinctive Ring/CWT 7	
Ring 8 Caller	Caller number pattern to play Distinctive Ring/CWT 8	
Default Ring	Default ringing pattern, 1 – 8, for all callers	1
Default CWT	Default CWT pattern, 1 – 8, for all callers	1
Hold Reminder Ring	Ring pattern for reminder of a holding call when the phone is on-hook	None
Call Back Ring	Ring pattern for call back notification	None
Cfwd Ring Splash Len[2]	Duration of ring splash when a call is forwarded (0 – 10.0s)	0
Cblk Ring Splash Len[2]	Duration of ring splash when a call is blocked (0 – 10.0s)	0
VMWI Ring Splash Len	Duration of ring splash when new messages arrive before the VMWI signal is applied (0 – 10.0s)	.5

Notes:
1. Caller number patterns are matched from Ring 1 to Ring 8. The first match (not the closest match) will be used for alerting the subscriber.
2. Feature not yet available.

3.4.7. Regional Parameters

3.4.7.1. Call Progress Tones

Parameter Name	Description	Default
Dial Tone[1]	Played when prompting the user to enter a phone number	350@-19,440@-19;10(*/0/1+2)
Second Dial Tone	An alternative to <Dial Tone> when user tries to dial a 3-way call	420@-19,520@-19;10(*/0/1+2)
Outside Dial Tone[1]	An alternative to <Dial Tone> usually used to prompt the user to enter an external phone number (versus an internal extension). This is triggered by a "," character encountered in the dial plan.	420@-16;10(*/0/1)
Prompt Tone[1]	Played when prompting the user to enter a call forward phone number	520@-19,620@-19;10(*/0/1+2)
Busy Tone	Played when a 486 RSC is received for an outbound call	480@-19,620@-19;10(.5/.5/1+2)
Reorder Tone[1,2]	Played when an outbound call has failed or after the far end hangs up during an established call	480@-19,620@-19;10(.25/.25/1+2)
Off Hook Warning Tone[2]	Played when the subscriber does not place the handset on the cradle properly	480@-10,620@0;10(.125/.125/1+2)
Ring Back Tone	Played for an outbound call when the far end is ringing	440@-19,480@-19;*(2/4/1+2)
Confirm Tone	This should be a brief tone to notify the user that the last input value has been accepted.	600@-16;1(.25/.25/1)"
SIT1 Tone	An alternative to <Reorder Tone> played when an error	985@-16,1428@-

	occurs while making an outbound call. The RSC to trigger this tone is configurable (see Section ???)	16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT2 Tone	See <SIT1 Tone>	914@-16,1371@-16,1777@-16;20(.274/0/1,.274/0/2,.380/0/3,0/4/0)
SIT3 Tone	See <SIT1 Tone>	914@-16,1371@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT4 Tone	See <SIT 1 Tone>	985@-16,1371@-16,1777@-16;20(.380/0/1,.274/0/2,.380/0/3,0/4/0)
MWI Dial Tone[1]	This tone is played instead of <Dial Tone> when there are unheard messages in the subscriber's mail box	350@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2)
Cfwd Dial Tone	Special dial tone played when call forward all is activated	350@-19,440@-19;2(.2/.2/1+2);10(*/0/1+2)
Holding Tone	Indicate to the local user that the far end has placed the call on hold	600@-16;*(.1/.1/1,.1/.1/1,.1/9.5/1)
Conference Tone	Plays to all parties when a 3-way conference is in progress	350@-16;30(.1/.1/1,.1/9.7/1)

Notes:
1. Reorder Tone is played automatically when <Dial Tone> or any of its alternatives times out
2. Off Hook Warning Tone is played when Reorder Tone times out

3.4.7.2. Ring and CWT Cadence

Parameter Name	Description	Default
Ring1 Cadence	Cadence script for distinctive ring 1	60(2/4)"
Ring2 Cadence	Cadence script for distinctive ring 2	60(.3/.2,1/.2,.3/4)"
Ring3 Cadence	Cadence script for distinctive ring 3	60(.8/.4,.8/4)
Ring4 Cadence	Cadence script for distinctive ring 4	60(.4/.2,.3/.2,.8/4)
Ring5 Cadence	Cadence script for distinctive ring 5	60(.4/.2,.3/.2,.8/4)
Ring6 Cadence	Cadence script for distinctive ring 6	60(.4/.2,.3/.2,.8/4)
Ring7 Cadence	Cadence script for distinctive ring 7	60(.4/.2,.3/.2,.8/4)
Ring8 Cadence	Cadence script for distinctive ring 8	60(.4/.2,.3/.2,.8/4)
CWT 1 Cadence	Cadence script for distinctive CWT 1	30(.3/9.7)
CWT2 Cadence	Cadence script for distinctive CWT 2	30(.1/.1, .1/9.7)"
CWT3 Cadence	Cadence script for distinctive CWT 3	30(.1/.1, .1/.1, .1/9.5)
CWT4 Cadence	Cadence script for distinctive CWT 4	30(.1/.1, .3/.1, .1/9.3)
CWT5 Cadence	Cadence script for distinctive CWT 5	30(.3/.1,.1/.1,.3/9.1)
CWT6 Cadence	Cadence script for distinctive CWT 6	30(.1/.1, .3/.1,

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		.1/9.3)
CWT7 Cadence	Cadence script for distinctive CWT 7	30(.1/.1, .3/.1, .1/9.3)
CWT8 Cadence	Cadence script for distinctive CWT 8	2.3(..3/2)
Ring1 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 1 for the inbound call	Bellcore-r1
Ring2 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 2 for the inbound call	Bellcore-r2
Ring3 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 3 for the inbound call	Bellcore-r3
Ring4 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 4 for the inbound call	Bellcore-r4
Ring5 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 5 for the inbound call	Bellcore-r5
Ring6 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 6 for the inbound call	Bellcore-r6
Ring7 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 7 for the inbound call	Bellcore-r7
Ring8 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 8 for the inbound call	Bellcore-r8
Ring Waveform	Waveform for the ringing signal	Sinusoid
Ring Frequency	Frequency of the ringing signal. Valid values are 10 – 100 (Hz)	25
Ring Voltage	Ringing voltage. 60-90 (V)	70
CWT Frequency	Frequency script of the call waiting tone. All distinctive CWT is based on this tone.	440@-10

Notes:

3.4.7.3. Control Timer Values (sec)

Parameter Name	Description	Default
Hook Flash Timer Min	Minimum on-hook time before off-hook to qualify as hook-flash. Less than this the on-hook event is ignored. Range: 0.1 – 0.4 sec	0.1
Hook Flash Timer Max	Maximum on-hook time before off-hook to qualify as hook-flash. More than this the on-hook event is treated as on-hook (no hook-flash event). Range: 0.4 – 1.6 sec	0.9
Callee On Hook Delay	The phone must be on-hook for at this time in sec before the SPA will tear down the current inbound call. It does not apply to outbound calls. Range: 0 – 255 sec	0
Reorder Delay	Delay after far end hangs up before reorder tone is played. 0 = plays immediately, inf = never plays. Range: 0 – 255 sec	5
Call Back Expires	Expiration time in sec of a call back activation. Ragne: 0 – 65535 sec	1800
Call Back Retry Intvl	Call back retry interval in sec. Range: 0 – 255 sec	30
Call Back Delay	Delay after receiving the first SIP 18x response before declaring the remote end is ringing. If a busy response is received during this time, the SPA still considers the call as failed and keeps on retrying.	0.5
VMWI Refresh Intvl	Interval between VMWI refresh to the CPE	0.5
Interdigit Long Timer[3]	Long timeout between entering digits when dialing. Range: 0 – 64 sec	10

Interdigit Short Timer[3]	Short timeout between entering digits when dialing. Range: 0 – 64 sec	3

Notes:
1. The Call Progress Tones and DTMF playback level are not affected by the <FXS Port Output Gain>.
2. The interdigit timer values are used as defaults when dialing. The Interdigit_Long_Timer is used after any one digit, if all valid matching sequences in the dial plan are incomplete as dialed. The Interdigit_Short_Timer is used after any one digit, if at least one matching sequence is complete as dialed, but more dialed digits would match other as yet incomplete sequences.

3.4.7.4. Vertical Service Code Assignment

Parameter Name	Description	Default
Call Return Code	Call the last caller.	*69
Blind Transfer Code	Blind transfer current call to the target specified after the activation code	*98
Cfwd All Act Code	Forward all calls to the target specified after the activation code	*72
Cfwd All Deact Code	Cancel call forward all	*73
Cfwd Busy Act Code	Forward busy calls to the target specified after the activation code	*90
Cfwd Busy Deact Code	Cancel call forward busy	*91
Cfwd No Ans Act Code	Forward no-answer calls to the target specified after the activation code	*92
Cfwd No Ans Deact Code	Cancel call forward no-answer	*93
Cfwd Last Act Code	Forward the last inbound or outbound calls to the target specified after the activation code	*63
Cfwd Last Deact Code	Cancel call forward last	*83
Block Last Act Code	Block the last inbound call	*60
Block Last Deact Code	Cancel blocking of the last inbound call	*80
Accept Last Act Code	Accept the last outbound call. Let it ring through when DND or Call Forward All is in effect	*64
Accept Last Deact Code	Cancel Accept Last	*84
Call Back Act Code	Callback when the last outbound call is not busy	*66
Call Back Deact Code	Cancel callback	*86
CW_Act_Code	Enable Call Waiting on all calls	*56
CW_Deact_Code	Disable Call Waiting on all calls	*57
CW_Per_Call_Act_Code	Enable Call Waiting for the next call	*71
CW_Per_Call_Deact_Code	Disable Call Waiting for the next call	*70
Block_CID_Act_Code	Block CID on all outbound calls	*67
Block_CID_Deact_Code	Unblock CID on all outbound calls	*66
Block_CID_Per_Call_Act_Code	Block CID on the next outbound call	*81
Blcok_CID_Per_Call_Deact_Code	Unblock CID on the next inbound call	*82
Block_ANC_Act_Code	Block all anonymous calls	*77
Block_ANC_Deact_Code	Unblock all anonymous calls	*87
DND_Act_Code	Enable Do Not Disturb	*78
DND_Deact_Code	Disable Do Not Disturb	*79
CID_Act_Code	Enable Caller-ID Generation	*65
CID_Deact_Code	Disable Call-ID Generation	*85
CWCID_Act_Code	Enable Call Waiting Caller-ID generation	*25
CWCID_Deact_Code	Disable Call Waiting Caller-ID generation	*45

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Dist_Ring_Act_Code	Enable Distinctive Ringing	*61
Dist_Ring_Deact_Code	Disable Distinctive Ringing	*81
Speed Dial Activation Code	Assign a speed dial number	*74

Notes:
1. These codes automatically appended to the dial-plan. So no need to include them in dial-plan (although no harm to do so either).

3.4.7.5. Miscellaneous Parameters

Parameter Name	Description	Default
Local Date (mm/dd/yyyy)	Setting the local date; year is optional and can be 2-digit or 4-digit	
Local Time (HH/mm/ss)	Setting the local time; second is optional.	
Time Zone	Number of hours to add to GMT to form local time for caller-id generation	GMT-07:00
FXS Port Impedance	Electrical impedance of the FXS port.	600
FXS Port Input Gain	Input Gain in dB. Valid values are 6.0 to –infinity. Up to 3 decimal places	-3
FXS Port Output Gain	Similar to <FXS Port Input Gain> but apply to the output signal	-3
DTMF Playback Level	Local DTMF playback level in dBm (up to 1 decimal place)	-10.0
DTMF Playback Length	Local DTMF playback duration in ms	.1
Detect ABCD	Enable local detection of DTMF ABCD	Yes
Playback ABCD	Enable local playback of OOB DTMF ABCD	Yes

3.5. Call Statistics Reporting

The following lists the statistics collected by the SPA during normal operation. These statistics are presented in the SPA web-page (under the “Info” tab). Line status is reported for each line (1 and 2). Each line maintains up to 2 calls: Call 1 and 2.

System Status	
Current Time	Current time and date. E.g., 10/3/2003 16:43:00
Elapsed Time	Total time elapsed since last reboot. E.g., 25 days and 18:12:36
Broadcast Pkts Sent	Total number of broadcast packets sent
Broadcast Pkts Recv	Total number of broadcast packets received
Broadcast Bytes Sent	Total number of broadcast bytes sent
Broadcast Bytes Recv	Total number of broadcast bytes received and processed
Broadcast Packets Dropped	Total number of broadcast packets received but not processed
Broadcast Bytes Dropped	Total number of broadcast bytes received but not processed
RTP Packets Sent	Total number of RTP packets sent (including redundant packets)
RTP Packets Received	Total number of RTP packets received (including redundant packets)
RTP Bytes Sent	Total number of RTP bytes sent
RTP Bytes Received	Total number of RTP bytes received
SIP Messages Sent	Total number of SIP messages sent (including retransmissions)
SIP Messages Received	Total number of SIP messages received (including retransmissions)
SIP Bytes Sent	Total number of bytes of SIP messages sent (including retransmissions)
SIP Bytes Received	Total number of bytes of SIP messages received (including retransmissions)
External IP	External IP address used for NAT mapping
Line 1/2 Status	
Hook State	State of the hook switch: On or Off
Registration State	Registration state of the line: Not Registered, Registered or Failed
Last Registration At	Local time of the last successful registration

Next Registration In	Number of seconds before the next registration renewal
Message Waiting	Indicate whether new voice mails available: Yes or No
Call Back Active	Indicate whether a call back request is in progress: Yes or No
Last Called Number	The last number called
Last Caller Number	The number of the last caller
Mapped SIP Port	NAT Mapped SIP Port
Call 1/2 Status	
State	State of the call: Idle, Dialing, Calling, Proceeding, Ringing, Answering, Connected, Hold, Holding, Resuming, or Reorder
Tone	Tone playing for this call: Dial, 2nd Dial, Outside Dial, Ring Back, Ring, Busy, Reorder, SIT1– 4, Call Waiting, Call Forward, Conference, Prompt, Confirmation, or Message-Waiting
Encoder	Encoder in use: G711u, G711a, G726-16/24/32/40, G729a, or G729ab
Decoder	Decoder in use: G711u, G711a, G726-16/24/32/40, G729a, or G729ab
FAX	Indicate whether FAX pass-through mode has been initiated: Yes or No
Type	Indicate the call type: Inbound or Outbound
Remote Hold	Indicate whether the remote end has placed the call on hold: Yes or No
Call Back	Indicate whether the call is triggered by a call back request: Yes or No
Peer Name	Name of the peer
Peer Phone	Phone number of the peer
Duration	Duration of the call in hr/min/sec format
Packets Sent	Number of RTP packets sent
Packets Recv	Number of RTP packets received
Bytes Sent	Number of RTP bytes sent
Bytes Recv	Number of RTP bytes received
Decode Latency	Decoder latency in milliseconds
Jitter	Receiver jitter in milliseconds
Frames Lost	Total number of frames lost in milliseconds
Packet Error	Number of RTP packets received that are invalid
Mapped RTP Port	NAT mapped RTP port

4. User Guidelines

The SPA can be configured to the custom requirements of the service provider Administrator, so that from the subscriber's point of view, the service behaves exactly as the service provider Administrator wishes – with varying degrees of control left with the end user. This means that a service provider can leverage the programmability of the SPA to offer sometimes subtle yet continually valuable and differentiated services optimized for the network environment or target market(s).

This section of the User Guide, describes how some of the supported basic and enhanced, or supplementary services could be implemented. The implementations described below by no means are the only way to achieve the desired service behavior. The specific feature activation and deactivation codes used in the below examples match the factory default values of the SPA-2000.

Please Note: *Refer to documentation provided by your service provider for the most accurate representation to the current service availability and access code values. Many of the below services – especially "enhanced services" – are subject to interoperation with the service provider's application/call server capabilities. Therefore some of the below services may not behave as described.*

To understand the specific implementation options of the below features, including parameters, requirements and contingencies please refer the section Configuration Parameters, section 3.4.

4.1. Basic Services

4.1.1. Originating a Phone Call

Service Description	Placing telephone a call to another telephone or telephony system (IVR, conference bridge, etc.). This is the most basic service.
User Action Required to Activate or Use	When the user picks up the handset, the SPA provides dial tone and is ready to collect dialing information via DTMF digits from the telephone touchtone key pad.
Expected Call and Network Behavior	While it is possible to support overlapped dialing within the context of SIP, the SPA collects a complete phone number and sends the full number in a SIP INVITE message to the proxy server for further call processing. In order to minimize dialing delay, the SPA maintains a dial plan and matches it against the cumulative number entered by the user. The SPA also detects invalid phone numbers not compatible with the dial plan and alerts the user via a configurable tone (Reorder).
User Action Required to Deactivate or End	Hang-up the telephone.

4.1.2. Receiving a Phone Call

Service Description	The SPA can receive calls from the PSTN or other IP Telephony subscribers
User Action Required to Activate or Use	When the telephone rings, pick up the handset and begin talking.
Expected Call and Network Behavior	Each subscriber is assigned an E.164 ID (phone number) so that they may be reached from wired or wireless callers on the PSTN or IP network. The SPA supplies ring voltage to the attached telephone set to alert the user of incoming calls.
User Action Required to Deactivate or End	Hang-up the telephone.

4.2. Enhanced Services

4.2.1. Caller ID

Service Description	If available, the SPA supports the generation and pass-through of Caller ID information.
User Action Required to Activate or Use	No user action required. The user's telephone equipment must support Caller ID to display

	the caller's name and/or number.
Expected Call and Network Behavior	In between ringing bursts, the SPA can generate a Caller-ID signal to the attached phone when the phone is on-hook. As part of the INVITE message, the SPA sends the caller's name and number as it is configured in the profile.
User Action Required to Deactivate or End	No user action required. See CLIP and CLIR.

4.2.2. Calling Line Identification Presentation (CLIP)

Service Description	Some users will elect to block their Caller ID information for all outgoing calls. However, there may be circumstances where sending Caller ID information for a call is desired, i.e. trying to reach a party that does not accept Caller ID blocked calls.
User Action Required to Activate or Use	Lift the receiver Listen for dial tone Press ***82** Listen for dial tone Dial the telephone number you are calling
Expected Call and Network Behavior	Caller ID will be sent to the distant party for this call only. Users must repeat this process at the start of each call.
User Action Required to Deactivate or End	No action required. This service is only in effect for the duration of the current call.

4.2.3. Calling Line Identification Restriction (CLIR) – Caller ID Blocking

Service Description	This feature allows the user to block the delivery of their Caller ID to the number they are calling.
User Action Required to Activate or Use	**To block Caller ID on the next outbound call, do the following:** Lift the receiver Listen for dial tone Press ***81** Listen for dial tone

	Dial the telephone number you are calling You must repeat this process at the start of each call. **To block Caller ID on all outbound calls, do the following:** Lift the receiver Listen for dial tone Press ***67** Listen for dial tone Dial the telephone number you are calling
Expected Call and Network Behavior	The user activates this service to hide his Caller ID when making an outgoing call.
User Action Required to Deactivate or End	If you used *81 to block Caller ID, no action required. This service is only in effect for the duration of the current call. If you used *67 to block Caller ID, do the following: Lift the receiver Listen for dial tone Press ***66** Listen for dial tone

4.2.4. Call Waiting

Service Description	The user can accept a call from a 3rd party while engaging in an active call. The SPA shall alert the subscriber of the 2nd incoming call by playing a call waiting tone.
User Action Required to Activate or Use	If you choose to answer the second call either: Press and release your phone's switch hook (the button you release when you take your phone off the hook) or: Press the “flash” button (if your phone has one). This puts your first call on hold and automatically connects you to your second call. To put your second caller back on hold and return to your first caller, press the switch hook or flash button again. (You can alternate between calls as often as you like.)

Expected Call and Network Behavior	If the user is on a call when another call comes in they will hear a series of beeps / tones alerting them to the second call. The person calling will hear normal ringing.
User Action Required to Deactivate or End	See Cancel Call Waiting.

4.2.5. Disable or Cancel Call Waiting

Service Description	The SPA supports disabling of call waiting permanently or on a per call basis.
User Action Required to Activate or Use	To temporarily disable Call Waiting (for the length of one call): Before placing a call: Lift Receiver Press ***70** Listen for dial tone then dial the number you want to call. Call Waiting is now disabled for the duration of this call only. To deactivate Call Waiting on a permanent basis (until cancelled): Lift the receiver Listen for dial tone Press ***57** You will hear a confirmation tone signaling your request to cancel Call Waiting has been accepted.
Expected Call and Network Behavior	Callers who dial your number will receive a busy signal or, if available, the caller will be forwarded to voice mail or another predetermined forwarding number.
User Action Required to Deactivate or End	If you have cancelled Call Waiting temporarily, no user action is required. If you deactivated call waiting and wish to reinstate the service, do the following: Lift the receiver Listen for dial tone Press ***56** You will hear a confirmation tone signaling your request to cancel Call Waiting has been

	accepted.

4.2.6. Call-Waiting with Caller ID

Service Description	When the user is on the phone and has Call Waiting active, the new caller's Caller ID information will be displayed on the users phone display screen at the same time the user is hearing the Call Waiting beeps / tones.
User Action Required to Activate or Use	The telephone equipment connected to the SPA must support Call-Waiting with Caller ID.
Expected Call and Network Behavior	In between call waiting tone bursts, the SPA can generate a Caller-ID signal to the attached phone when it is off hook.
User Action Required to Deactivate or End	Not applicable.

4.2.7. Voice Mail

Service Description	Service Providers may provide voice mail service to their subscribers. Users have the ability to retrieve voice mail via the telephone connected to the SPA.
User Action Required to Activate or Use	The SPA indicates that a message is waiting by, playing stuttered dial tone when the user picks up the handset. To retrieve messages: Lift the receiver Listen for dial tone Dial the phone number assigned to the SPA You will be connected to the voice mail server and prompted by a voice response system with instructions to listen to your messages.
Expected Call and Network Behavior	When voice mail is available for a subscriber, a notification message will be sent from the Voice Mail server to the SPA. When the user dials their own phone number, the SPA connects the subscriber their voice mail system which can then connect them to their individual voice mail box.
User Action Required to Deactivate or End	Follow instructions of the voice mail system or simply hang-up the telephone.

4.2.8. Attendant Call Transfer

Service Description	Attendant Call Transfer lets a customer use their touchtone phone to send a call to any other phone, inside or outside their business, including a wireless phones.
User Action Required to Activate or Use	While in a call with the party to be transferred: Press the switch hook or flash button on the phone to place the party on hold Listen for three short tones followed by dial tone Dial the number to which you will transfer the caller Stay on the line until the called number answers Announce the call Press the switch hook or flash button adding the held party to the call Hang up to connect the two parties and transfer the call Note: You can hook flash while the 3rd party is ringing to start an early conference. Then hang up to complete the transfer without waiting for the 3rd party to answer first.
Expected Call and Network Behavior	When the user presses the switch hook or flash button, the transferee is placed on hold. When the user successfully dials the transfer number and the party answers the transferee can be added to the call by pressing the switch hook or flash button creating a three-way conference. When the user hangs up the phone the transferee and the called party remain in a call.
User Action Required to Deactivate or End	Not applicable.

4.2.9. Unattended or “Blind” Call Transfer

Service Description	Unattended or “Blind” Call Transfer lets a customer use their Touchtone phone to send a call to any other phone, inside or outside their business, including a wireless phones.
User Action Required to Activate or Use	While in a call with the party to be transferred:

	Press the switch hook or flash button on the phone to place the party on hold Enter ***98** Dial the number to which you will transfer the caller The call is transferred when a complete number is entered. You will hear a short confirmation tone, followed by regular dial tone
Expected Call and Network Behavior	When the user presses the switch hook or flash button, the transferee is placed on hold. When the user successfully dials the transfer number, the transferee will automatically call the dialed number.
User Action Required to Deactivate or End	No applicable.

4.2.10. Call Hold

Service Description	Call Hold lets you put a caller on hold for an unlimited period of time. It is especially useful on phones without the hold button. Unlike a hold button, this feature provides access to a dial tone while the call is being held.
User Action Required to Activate or Use	Press the switch hook or flash button on the phone to place the first party on hold. You will hear a dial tone. To make another call: Enter the new number To return to call on hold: Hang up and the phone set will ring with the first call on the line (or press Hook Flash again)
Expected Call and Network Behavior	
User Action Required to Deactivate or End	Hang-up the telephone.

4.2.11. Three-Way Calling

Service Description	The user can originate a call to a 3rd party while engaging in an active call.
User Action Required to Activate or Use	Press the switch hook or flash button on the phone to place the first party on hold Listen for dial tone

	Dial the number of the 3rd party. When the 3rd party answers you may have a conversation with them while the other party is on hold. To hold a conference with the party on hold and the 3rd party, simply press the switch hook or flash button
Expected Call and Network Behavior	The SPA supports up to two calls per line. The SPA can conference two calls by bridging the 2nd and 3rd parties.
User Action Required to Deactivate or End	Hang-up the telephone.

4.2.12. Three-Way Ad-Hoc Conference Calling

Service Description	This feature allows the user to conference up to two other numbers on the same line to create a three-way call.
User Action Required to Activate or Use	If you are already on a call and wish to add a third party: Press the switch hook or flash button Listen for dial tone Dial the third party normally When the third party number starts to ring press the switch hook or flash button again You now have the original caller and the third party together with you on the same call. If you want to initiate a new Three Way Call: Call the first party in the normal manner Follow the directions for adding a third party (see instructions above)
Expected Call and Network Behavior	The SPA can host a 3-way conference and perform 3-way audio mixing (without the need of an external conference bridge device or service). If you also have Call Transfer you can also hang up at any time to transfer the original caller to the third party
User Action Required to Deactivate or End	

4.2.13. Call Return

Service Description	The SPA supports a service that allows the SPA to automatically dial the last caller's number.
User Action Required to Activate or Use	Pick up the receiver Listen for dial tone Press ***69** to dial back the last caller that tried to reach you.
Expected Call and Network Behavior	This service gives the user the convenience of recalling the last incoming call to their number automatically.
User Action Required to Deactivate or End	No user action required

4.2.14. Automatic Call Back

Service Description	This feature allows the user to place a call to the last number they tried to reach whether the call was answered or unanswered (not busy) by dialing an activation code.
User Action Required to Activate or Use	Pick up the receiver Listen for dial tone Press ***66**
Expected Call and Network Behavior	If the number called is idle the call will ring through and complete normally. If the called number is busy the SPA will monitor the called number for up to 30 minutes. When both lines are idle, the user hears a special ring. During the monitoring process the user can continue to originate and receive calls without affecting the Call Return on Busy request. Call Return on Busy requests can be canceled by dialing the deactivation code.
User Action Required to Deactivate or End	Lift the receiver Listen for dial tone Press ***86**

4.2.15. Call FWD – Unconditional

Service Description	All calls are immediately forwarded to the designated forwarding number. The SPA will not ring or provide call waiting when Call FWD

	– Unconditional is activated. The SPA can be configured to provide a short “half” ring or ring “splash” as a reminder that the SPA is unconditionally forwarding all calls.
User Action Required to Activate or Use	Lift the receiver Listen for dial tone Press *72 Listen for dial tone and enter the telephone number you are forwarding your call to. Activation will be confirmed with two beeps and short bursts of dial tone to alert you that your forwarding has been activated. Alternatively, the user can activate this feature from a web browser interface.
Expected Call and Network Behavior	This feature allows a user the option to divert (forward) all calls to their telephone number to any number using the touchtone keypad of their telephone or web browser interface. This service is activated or deactivated from the phone being forwarded or the web browser interface. While the Call Forward – Unconditional feature is active, the SPA will use a stuttered dial tone in addition to the optional ring splash to remind the user that calls are being unconditionally forwarded.
User Action Required to Deactivate or End	Lift the receiver Listen for dial tone Press *73 You will hear a dial tone signaling your change has been accepted. Alternatively, the user can deactivate this feature from a web browser interface.

4.2.16. Call FWD – Busy

Service Description	Calls are forwarded to the designated forwarding number if the subscriber’s line is busy because of the following; Primary line already in a call, primary and secondary line in a call or conference.
User Action Required to Activate or Use	Lift the receiver

	Listen for dial tone Press ***90** Listen for dial tone and enter the telephone number you are forwarding your call to. Activation will be confirmed with two beeps and then dial tone to alert you that your forwarding has been activated. Alternatively, the user can activate this feature from a web browser interface.
Expected Call and Network Behavior	This feature allows a user the option to divert (forward) calls to their telephone number to any number when their phone is busy or in conference by using the touchtone keypad of their telephone or web browser interface. This service is activated or deactivated from the phone being forwarded or the web browser interface.
User Action Required to Deactivate or End	Lift the receiver Listen for dial tone Press ***91** You will hear dial tone signaling your change has been accepted. Alternatively, the user can deactivate this feature from a web browser interface.

4.2.17. Call FWD - No Answer

Service Description	Calls are forwarded to the designated forwarding number after a configurable time period elapses while the SPA is ringing and does not answer.
User Action Required to Activate or Use	Lift the receiver Listen for dial tone Press ***92** Listen for dial tone and enter the telephone number you are forwarding your call to. Activation will be confirmed with two beeps and then dial tone to alert you that your forwarding has been activated. Alternatively, the user can activate this feature from a web browser interface. Note: The forward delay is entered from the

	web interface. The default forward delay is 20 seconds.
Expected Call and Network Behavior	This feature allows a user the option to divert (forward) calls to their telephone number to any other dialable number when their phone is not answered by using the touchtone keypad of their telephone or web browser interface. This service is activated or deactivated from the phone being forwarded or the web browser interface.
User Action Required to Deactivate or End	Lift the receiver Listen for dial tone Press ***93** You will hear dial tone signaling your change has been accepted. Alternatively, the user can deactivate this feature from a web browser interface.

4.2.18. Anonymous Call Blocking

Service Description	By setting the corresponding configuration parameter on the SPA, the subscriber has the option to block incoming calls that do not reveal the caller's Caller ID.
User Action Required to Activate or Use	Pick up the receiver Listen for dial tone To Activate Press ***77**
Expected Call and Network Behavior	When activated by the user, callers will hear (busy) tone.
User Action Required to Deactivate or End	To De-activate Press ***87**

4.2.19. Distinctive / Priority Ringing and Call Waiting Tone

Service Description	The SPA supports a number of ringing and call waiting tone patterns to be played when incoming calls arrive. The choice of alerting pattern to use is carried in the incoming SIP INVITE message inserted by the SIP Proxy Server (or other intermediate application server in the Service Provider's domain).
User Action Required to Activate or Use	Pick up the receiver

	Listen for dial tone Press ***61**
Expected Call and Network Behavior	With this service, incoming calls from up to __ telephone numbers can be automatically identified by distinctive ringing. A distinctive ringing pattern (i.e. short-long-short) accompanies incoming calls from the designated telephone numbers. If the user is engaged in conversation and a call from one of the designated numbers arrives, a distinctive call waiting tone (i.e. short-long-short) accompanies the incoming call. Calls from other telephone numbers ring normally.
User Action Required to Deactivate or End	To De-activate Press ***81**

4.2.20. Speed Calling – Up to Eight (8) Numbers or IP Addresses

Service Description	The SPA supports user programming of up to 8 long distance, local, international or emergency numbers and/or IP addresses for fast and easy access.
User Action Required to Activate or Use	Pick up the receiver Listen for dial tone Press ***74** Dial the single digit code under which the number is to be stored (2-9) Dial the complete number to be stored just as if you were going to dial it yourself Listen for Confirmation tone (two short beeps) Hang up or repeat the sequence Note: To enter IP addresses, a graphical user interface like a web browser must be used.
Expected Call and Network Behavior	Pick up the receiver Listen for dial tone Press single digit code assigned to the stored number (2-9) Press # to signal dialing complete The number is automatically dialed normally.
User Action Required to Deactivate or End	None

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5. Where to Get Support:

For support questions related to your Sipura Technology product, please contact the company from whom you purchased the SPA-2000 or an authorized Sipura Technology reseller.

Other Ways to Contact Sipura Technology Technical Support:
Partners, Resellers and Distributors of Sipura Technology products may use the below information to contact Sipura support.

Email: support@sipura.com
Fax: 408-572-5671

General Technical Support Hours: 9:00am to 5:00pm PST – Monday through Friday

6. Appendix I – Dial Plan Administration:

6.1.1. Dial Plan

The SPA-2000 allows each line to be configured with a distinct dial plan. The dial plan specifies how to interpret digit sequences dialed by the user, and how to convert those sequences into an outbound dial string.

The SPA-2000 syntax for the dial plan closely resembles the corresponding syntax specified by MGCP and MEGACO. Some extensions are added that are useful in an end-point.

The dial plan functionality is regulated by the following configurable parameters:

- Interdigit_Long_Timer
- Interdigit_Short_Timer
- Dial_Plan ([1] and [2])

Other timers are configurable via parameters, but do not directly pertain to the dial plan itself. They are discussed elsewhere in this document.

Interdigit Long Timer:

```
ParName:     Interdigit_Long_Timer

Default:     10
```

The Interdigit_Long_Timer specifies the default maximum time (in seconds) allowed between dialed digits, when no candidate digit sequence is as yet complete (see discussion of Dial_Plan parameter for an explanation of candidate digit sequences).

Interdigit Short Timer:

```
ParName:     Interdigit_Short_Timer

Default:     3
```

The Interdigit_Short_Timer specifies the default maximum time (in seconds) allowed between dialed digits, when at least one candidate digit sequence is complete as dialed (see discussion of Dial_Plan parameter for an explanation of candidate digit sequences).

Dial Plan[1] and Dial Plan[2]:

```
ParName:     Dial_Plan[1] and Dial_Plan[2]

Default:     ( *xx | [3469]11 | 0 | 00 | <:1408>[2-9]xxxxxx |
               1[2-9]xx[2-9]xxxxxx | 011x. )
```

The Dial_Plan parameters contain the actual dial plan scripts for each of lines 1 and 2.

Dial Plan Digit Sequences:

The plans contain a series of digit sequences, separated by the '|' character. The collection of sequences is enclosed in parentheses, '(' and ')'.

When a user dials a series of digits, each sequence in the dial plan is tested as a possible match. The matching sequences form a set of candidate digit sequences. As more digits are entered by the user, the set of candidates diminishes until only one or none are valid.

Any one of a set of terminating events triggers the SPA-2000 to either accept the user-dialed sequence, and transmit it to initiate a call, or else reject it as invalid. The terminating events are:

- No candidate sequences remain: the number is rejected.
- Only one candidate sequence remains, and it has been matched completely: the number is accepted and transmitted after any transformations indicated by the dial plan, unless the sequence is barred by the dial plan (barring is discussed later), in which case the number is rejected.
- A timeout occurs: the digit sequence is accepted and transmitted as dialed if incomplete, or transformed as per the dial plan if complete.
- An explicit 'send' (user presses the '#' key): the digit sequence is accepted and transmitted as dialed if incomplete, or transformed as per the dial plan if complete.

The timeout duration depends on the matching state. If no candidate sequences are as yet complete (as dialed), the Interdigit_Long_Timeout applies. If a candidate sequence is complete, but there exists one or more incomplete candidates, then the Interdigit_Short_Timeout applies.

White space is ignored, and may be used for readability.

Digit Sequence Syntax:

Each digit sequence within the dial plan consists of a series of elements, which are individually matched to the keys pressed by the user. Elements can be one of the following:

- Individual keys '0', '1', '2' . . . '9', '*', '#'.
- The letter 'x' matches any one numeric digit ('0' .. '9')
- A subset of keys within brackets (allows ranges): '[' set ']' (e.g. [389] means '3' or '8' or '9')
    - Numeric ranges are allowed within the brackets: digit '-' digit (e.g. [2-9] means '2' or '3' or … or '9')
    - Ranges can be combined with other keys: e.g. [235-8*] means '2' or '3' or '5' or '6' or '7' or '8' or '*'.

Element repetition:

Any element can be repeated zero or more times by appending a period ('.' character) to the element. Hence, "01." matches "0", "01", "011", "0111", … etc.

Subsequence Substitution:

A subsequence of keys (possibly empty) can be automatically replaced with a different subsequence using an angle bracket notation: '<' dialed-subsequence ':' transmitted-subsequence '>'. So, for example, "<8:1650>xxxxxxx" would match "85551212" and transmit "16505551212".

Intersequence Tones:

An “outside line” dial tone can be generated within a sequence by appending a ‘,’ character between digits. Thus, the sequence “9, 1xxxxxxxxx” sounds an “outside line” dial tone after the user presses ‘9’, until the ‘1’ is pressed.

Number Barring:

A sequence can be barred (rejected) by placing a ‘!’ character at the end of the sequence. Thus, “1900xxxxxxx!” automatically rejects all 900 area code numbers from being dialed.

Interdigit Timer Master Override:

The long and short interdigit timers can be changed in the dial plan (affecting a specific line) by preceding the entire plan with the following syntax:

- Long interdigit timer: ‘L’ ‘:’ delay-value ‘,’
- Short interdigit timer: ‘S’ ‘:’ delay-value ‘,’

Thus, “L=8,(. . .)” would set the interdigit long timeout to 8 seconds for the line associated with this dial plan. And, “L:8,S:4,(. . .)” would override both the long and the short timeout values.

Local Timer Overrides:

The long and short timeout values can be changed for a particular sequence starting at a particular point in the sequence. The syntax for long timer override is: ‘L’ delay-value ‘ ‘. Note the terminating space character. The specified delay-value is measured in seconds. Similarly, to change the short timer override, use: ‘S’ delay-value <space>.

Pause:

A sequence may require an explicit pause of some duration before continuing to dial digits, in order for the sequence to match. The syntax for this is similar to the timer override syntax: ‘P’ delay-value <space>. The delay-value is measured in seconds.

This syntax allows for the implementation of Hot-Line and Warm-Line services. To achieve this, one sequence in the plan must start with a pause, with a 0 delay for a Hot Line, and a non-zero delay for a Warm Line.

Implicit sequences:

The SPA-2000 implicitly appends the vertical code sequences entered in the Regional parameter settings to the end of the dial plan for both line 1 and line 2. Likewise, if Enable_IP_Dialing is enabled, then ip dialing is also accepted on the associated line.

Examples:

The following dial plan accepts only US-style 1 + area-code + local-number, with no restrictions on the area code and number.

```
( 1 xxx xxxxxxx )
```

The following also allows 7-digit US-style dialing, and automatically inserts a 1 + 212 (local area code) in the transmitted number.

```
( 1 xxx xxxxxxx | <:1212> xxxxxxx )
```

For an office environment, the following plan requires a user to dial 8 as a prefix for local calls and 9 as a prefix for long distance. In either case, an “outside line” tone is played after the initial 8 or 9, and neither prefix is transmitted when initiating the call.

```
( <9,:> 1 xxx xxxxxxx | <8,:1212> xxxxxxx )
```

The following allows only placing international calls (011 call), with an arbitrary number of digits past a required 5 digit minimum, and also allows calling an international call operator (00). In addition, it lengthens the default short interdigit timeout to 4 seconds.

```
S:4, ( 00 | 011 xxxxx x. )
```

The following allows only US-style 1 + area-code + local-number, but disallows area codes and local numbers starting with 0 or 1. It also allows 411, 911, and operator calls (0).

```
( 0 | [49]11 | 1 [2-9]xx [2-9]xxxxxx )
```

The following allows US-style long distance, but blocks 9xx area codes.

```
( 1 [2-8]xx [2-9]xxxxxx )
```

The following allows arbitrary long distance dialing, but explicitly blocks the 947 area code.

```
( 1 947 xxxxxxx ! | 1 xxx xxxxxxx )
```

The following implements a Hot Line phone, which automatically calls 1 212 5551234.

```
( S0 <:12125551234> )
```

The following provides a Warm Line to a local office operator (1000) after 5 seconds, unless a 4 digit extension is dialed by the user.

```
( P5 <:1000> | xxxx )
```

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